#5
8-21-03

PATENT APPLICATION
Attorney Doc. No. 2705-90

In re application of: Ilya Umansky, Neil Joffe, Shamin Sharifuddin Pirzada, and Dhaval N. Shah

Serial No. 09/492,423 Examiner: Hsu, Alpus

Confirmation No. 9857

Filed: January 27, 2000 Group Art Unit: 2665

For: **VOICE OVER INTERNET PROTOCOL CALL FALLBACK FOR QUALITY OF SERVICE DEGRADATION**

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

DECLARATIONS OF NEIL JOFFE AND DHAVAL N. SHAH

UNDER RULE 37 C.F.R. 1.131

We, Neil Joffe and Dhaval N. Shah, declare the following:

1. We are two of the co-inventors of the subject matter described in the present U.S. pending patent application entitled: VOICE OVER INTERNET PROTOCOL CALL FALLBACK FOR QUALITY OF SERVICE DEGRADATION, U.S. Serial No. 09/492,423, filed January 27, 2000.

2. We currently work for Cisco Systems, Inc. Our work mailing address is 170 West Tasman Drive, San Jose, CA 95134-1706.

3. Before September 30, 1999, we and the other named inventors of U.S. Serial No. 09/492,423 conceived of the idea of establishing a Voice over IP (VoIP) call over a VoIP

network and monitoring the quality of service on the VoIP network during the VoIP call. We came up with the idea of setting up a fallback call over a circuit switched network during the VoIP call when the monitored quality of service of the VoIP network degrades. We also conceived of the idea of establishing the circuit switched fallback call mid-call of the VoIP call where both the circuit switched connection and the VoIP connection would both carry the same call until the incoming call has completed redirection over the circuit switched connection. If the VoIP network conditions improve, the call could be rerouted back over a new VoIP connection.

4. Attached as Exhibit A is a disclosure document that was written prior to September 30, 1999, where we and the other named inventors of the '423 patent application describe our invention. The invention disclosure form was submitted to our employer, Cisco Systems. The Invention Disclosure Form describes how a Quality of Service (QoS) is performed for an in-progress Voice Over IP (VoIP) call. If congestion is detected for the in-progress VoIP call, then a PSTN fallback call is established and the receiving gateway bridges the PSTN call with the currently established VoIP call. Thus, both the receiving gateway for a time receives the same call over the PSTN fallback connection while continuing to receive packets for the same call over the VoIP connection. A VoIP connection can be reestablished and the call switched from the PSTN connection back to the VoIP connection when the QoS conditions improve for the VoIP network.

5. The Thornton et al. reference (US 6,363,065) and the Wellard et al. reference (US 6,510,219) were used to reject the claims of our application under 35 U.S.C. § 102(e). Applicants wish to "swear behind" these reference. Although Thornton et al has an effective filing date of November 10, 1999 and Wellard has an effective filing date of September 30, 1999 that precede the application's effective filing date of January 27, 2000, applicants conceived of their invention prior to September 30, 1999 and were diligent in reducing the concept to practice up until the time the patent application was filed on January 27, 2000.

I, the undersigned, declare that all statements made herein of my own knowledge are true, and that all statements made on information and belief are believed to be true; and further, that these statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United Stated Code, and that such willful false statements may jeopardize the validity of the application of any patent issuing thereon.

DATED this 11th day of August, 2003.

Neil Joffe

I, the undersigned, declare that all statements made herein of my own knowledge are true, and that all statements made on information and belief are believed to be true; and further, that these statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United Stated Code, and that such willful false statements may jeopardize the validity of the application of any patent issuing thereon.

DATED this 11th day of August, 2003.

DN Shah
Dhaval N. Shah

Patent Idea Details for Idea #51626

GENERAL INFORMATION

Title: VoIP call PSTN fall-back in case of QoS degradatioin

ID: 51626

Patent No.: —, ---

URL: [Application No. ---]

Inventors: Ilya Umansky (ilyau) and Neil Joffe (njoffe)
More details on these inventors listed below.

Date Entered: [redacted]

Date Modified: [redacted]

Date Filed: ---

Date Issued: ---

Background: During a voip call ip network congestion could degrade QoS of the call in-progress. PSTN fall-back of the ongoing call would be required for the call to continue.

Summary: The patent allows to perform the PSTN fall-back seamlessly, without interrupting the call in-progress. It also allows to switch from PSTN back to voip when the network condition improves without interrupting the ongoing call. Switching from/to PSTN/voip during a call any number of times is possible.

When either: sending or receiving gateway detects unacceptable degradation of QoS of a call in-progress it triggers PSTN fall-back.

Consider first a sending gateway which detects the call QoS degradation. The sending gateway places a PSTN call to the destination gateway. If sending or receiving gateway have CAS connection sending gateway sends DTMFs to signal to the receiving gateway information about the session to fall-back. Receiving gateway bridges(rather conferences) the PSTN DS0 to the outgoing off-ramp DS0 while continuing to receive voip packets for this session and sending them to the outgoing off-ramp DS0.

It then answers the call. Once sending gateway received call answer it cross-connects the incoming DS0 to the outgoing to receiving gateway DS0 and the call continues over the PSTN bypass. Alternatively in case of ISDN presence between the sending and receiving gateways, sending gateway instead of cross-connecting incoming DS0 to outgoing DS0 could continue to packetize incoming voice and send the voice packets over ISDN bypass to the receiving gateway. In this case one ISDN bypass could carry traffic for packets from multiple calls(likely up to 6 depending on the codec used). Each packet identifies what call it belongs to.
Thus, in case of bypass over ISDN first the gateway checks if there is already an ISDN connection to the receiving gateway carrying other calls. If it exists and bandwidth allows this connection could be used to carry this call. Otherwise a new connection is established.

Receiving gateway could detect a QoS degradation. It acts similarly to the above: establishes a PSTN connection to the sending gateway, signals what session the connection concerns(this is not necessary if voice will be sent in packets) and sending gateway routes calls over this PSTN connection.

Once network condition is improved call(s) carried by PSTN bypass could be seamlessly rerouted over ip again and PSTN connection is teared down.

This contributes and simplifies a new call addmission controll.
A new call is not to be addmited by the onramp gateway if there are calls in progress bypassing by PSTN from this onramp gateway to the same offramp gateway as a new call.

Advantages: Guaranteed call completion.
Transparent for user swich back and forth to/from ip/PSTN.
Optimized use of PSTN and switch back to ip as early as ip congestion is over.

No dependancy on telephony protocols to transfer calls: bridging is done on the router.

Cisco Use: VoIP gateways.

Method of code inspection, network monitoring.

EXH A

PAGE 2 OF 2